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(54) Sound environment simulator and a method of analyzing a sound space

Schallumgebungsimulator und Verfahren zur Schallfeldanalyse

Simulateur de son dans un environnement et méthode pour l'analyse de l'espace sonore

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(56) References cited:
EP-A- 0 432 973 US-A- 4 928 250

- PATENT ABSTRACTS OF JAPAN vol. 13, no. 395 (P-927) 04 September 1989 & JP-A-01 141 321 (MATSUSHITA) 02 June 1989
- PATENT ABSTRACTS OF JAPAN vol. 15, no. 176 (P-1198) 07 May 1991 & JP-A-03 038 695 (SHIMIZU) 19 February 1991

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Description

BACKGROUND OF THE INVENTION

(1) Field of the Invention

[0001] The present invention relates to a sound environment simulator which enables one to experience a computer simulation of sound performance when modeling acoustic affects in a listening room, a concert hall or the like, and to a method of analyzing spatial acoustic characteristics of a sound space.

(2) Description of the Related Art

[0002] Rapid advancement in the field of audio-visual technology makes it possible to supply high-quality music and images to individual homes. In particular, what has been attracting people's interest is a technique, a so-called "home theater", by which one can enjoy a movie or music at home as if he were in a movie theater or a concert hall. To enjoy the home theater, it is desirable to have a room where acoustic affects and sound insulation effects are well taken into account. However, in most of the cases, people enjoy movies and music in their living rooms where a variety of furniture and household appliances are installed. Thus, nowadays, not only the appearance and the feeling of being comfortable, but also the acoustic affects and insulation effects are important elements when designing for living rooms.

[0003] Conventionally, data related to acoustic characteristics are evaluated for modeling acoustic affects: they are computed by either experimental or analytical method and then fed back to design specifications.

[0004] In the experimental method, a model room is constructed, and an impulse response is measured directly to compute the acoustic characteristics data of the model room. However, the costs both for building materials and labor do not allow its application to individual house construction.

[0005] Whereas in the analytical method, the acoustic characteristics data are computed by a computer as it simulates a three dimensional sound field. This method has been widely used in recent years. Because the resulting data reliably correspond to various input conditions, become a reliable source for further analysis, and facilitate further processing and feedback to the design specifications.

[0006] Data necessary for simulating the three dimensional sound field are computed either by a numerical analysis or a geometric analysis based on theory of acoustics: sound waves are taken into account in the former, whereas they are not in the latter.

[0007] A finite element method(FEM) and a boundary element method(BEM) are typically used in the numerical analysis. In both the methods, the computer computes acoustic intrinsic modes or pressure distribution of a sound by solving a dominance equation of a stationary sound field, or namely Helmholtz's equation. Taking the sound waves into account makes it possible to analyze diffraction and interference of the sound; however, it does not allow the computation of a non-stationary value such as an echotime pattern. In particular, to compute the echotime pattern of a high frequency sound, a myriad number of lattices are involved, so that not only it takes relatively long, but also the amount thereof exceeds a memory capacity. For this reason, the numerical analysis has not been applied to practical applications.

[0008] On the other hand, in the geometric analysis based on the theory of acoustics, methods using sound rays or virtual images are typically used on the premise that the sound rays are reflected geometrically from walls. These methods are practical; for the memory capacity does not limit the amount of the computation. However, this analysis does not guarantee accuracy in all cases. For example, the resulting data may not be accurate when simulating the sound field with a low frequency sound in a room where a wave length is longer than its size. Because given these circumstances, the sound rays fail to reflect in accordance with the premise.

[0009] In addition, a reverberation time, which is a critical element to compute the acoustic characteristics data, is approximated by Sabine's or Eyring's equation to minimize the amount of computation for the sound ray tracing. Such an approximation causes an computation error when this analysis is applied to a space where a variety of household appliances are to be installed or whose shape is complex.

[0010] Another type of sound environment simulator has been developed in recent years. This type of simulator simulates a sound field by reproducing a sound in accordance with the acoustic characteristics data, so that one can hear the reproduced sound. In a broad sense, the sound environment simulator includes a model room where one can listen to a sound reproduced over speakers installed therein. However, this type of simulator is not applicable for individual house construction.

[0011] Accordingly, a sound environment simulator, disclosed in U.S. patent application No. 4,731,848, was proposed. This simulator is designed to reproduce a sound over a headphone with reverberation effects. More precisely, an input audio signal is branched and the branched signal is inputted into a delay circuit. Then, the delay circuit delays the output of the branched signal for latency comparable to the time difference of the reflected sound to produce a reverberant signal. Subsequently, the original input audio signal and reverberant signal are composited to generate an output signal which is transduced into a sound. However, the sound reproduced in this way may give an impression to a listener that it does not sound realistic; for the sounds of higher order reflections (i.e. the sounds reflected for a number of times) are discarded.

[0012] Moreover, this type of simulator does not re-

produce the sound immediately. This is because index data such as locations of a virtual sound source and the listener are inputted manually each time they move. Furthermore, the shape of a space selected as a model environment is limited.

[0013] US-A-4 928 250 discloses a system for deriving radiation images of an environment with an algorithm using form factors for setting analysis conditions. To find the form factors, a hemi-cube is constructed around the surface with grid cells defined for all faces on the hemi-cube. The identity of the polygons in the environment are stored to speed up the calculations. While US-A-4 928 250 mentions the possibility of using its invention for sound, US-A-4 928 250 discloses only radiation images of light in details and does not disclose how to apply it to sound. Since light radiation does not involve noticeable time delay, US-A-4 928 250 does not disclose or suggest use of an echotime pattern which represents time delays of radiation reverberation.

[0014] JP-A-11 041 321 discloses an audible type sound field simulator which uses a signal into a reverberation adding device 14 branched from the sound source music 9 to produce reverberation sound 15. The reverberation sound is added to music data 11, 12 to obtain music data 16, 17. This device uses an echo time pattern obtained by acoustic wave simulation by a virtual image method, and does not use a geometric method based on the theory of acoustics.

[0015] JP-A-30 038 695 discloses an audible in-room sound field simulator comprising an initial reflected sound generation part, an initial scattered sound generation part and a reverberation generation part. The initial reflect sound I, initial scattered sound S, reverberations R are generated separately by the respective parts of the simulator. This simulator does not use an echo time pattern to compute reverberation sound or initial scattered sound.

SUMMARY OF THE INVENTION

[0016] Accordingly, the present invention has an object to provide a sound environment simulator which enables a listener to experience a simulation of sound environment of a space selected as a model environment without constructing the model space.

[0017] According to an aspect of the present invention, there is provided a sound environment simulator which comprises echotime pattern computing means for computing an echotime pattern of a space; impulse response computing means for computing an impulse response using said echotime pattern; reproduced sound generating means for generating reproduced sound data using said impulse response; and output means for outputting a sound by transducing said reproduced sound data into the sound. The present simulator is characterised in that said echotime pattern computing means is adapted to compute said echotime pattern by a geometric method based on the theory of acoustics with input data as to conditions of said space; said impulse response computing means is adapted to compute a response for each frequency band from said echotime pattern and to compute a composite response to generate said impulse response; and said reproduced sound generating means is adapted to generate the reproduced sound data by convolution of said impulse response on non-reverberate sound source data.

[0018] The simulator may further comprise analysis data setting means for setting analysis data by receiving analysis condition data including a shape and physical data of walls of said space, dividing said walls into a set of certain sections, and setting a virtual sound source and a virtual sound receiving point at respective arbitrary positions within said space. Said echotime pattern means may be adapted to receive said setting analysis data and include time series data computing means for computing, based cn said setting analysis data, time series data related to arrival volume of a sound emanated from said virtual sound source to said virtual sound receiving point; and data transducing means for transducing said time series data related to the arrival volume into said echotime pattern by interpolating certain data.

[0019] The simulator may further comprise dry source input means for inputting said non-reverberate sound source data; and head transfer function storing means for storing data related to a head transfer function; wherein said reproduced sound generating means is adapted to generate the reproduced sound data by convoluting said impulse response received from said impulse response computing means and said head transfer function received from said head transfer function storing means on said non-reverberate sound scurce data received from said dry source input means.

[0020] The simulator may further comprise location data detecting means for detecting data related to at least one of a listener's head location and a listener's head direction; wherein said reproduced sound generating means is adapted to extract said impulse response and said head transfer function corresponding to said detected location data to convolute them on said non-reverberate sound source data.

[0021] The location data detecting means may be adapted to detect time varying data. The output means may comprise a headphone.

[0022] The simulator may also comprise head transfer function storing means for storing data related to a head transfer function; and composite response generating means for computing a composite response for each direction of a listener's head by convolution of said impulse response and said head transfer function, wherein said reproduced sound generating means is adapted to generate reproduced sound data by convolution of said composite response on the non-reverberate sound source data.

[0023] The time series data computing means may be adapted to compute form factors to compute the arrival volume of the sound to the sound receiving point by us-

ing said form factors and a sound absorption coefficient of each section included in said physical data of the walls.

[0024] According to another aspect of the present invention, there is provided a methcd of analysing a sound space, comprising the steps of computing an echotime pattern of a space; computing an impulse response using said echotime pattern; generating reproduced sound data related to a reproduced sound based on said impulse response; and outputting a sound by transducing said reproduced sound data in the sound. The method is characterised by the step of detecting data related to a listener's location; wherein the step of computing an echotime pattern computes said echotime pattern for a virtual sound source placed arbitrarily in the space by a geometric method based on the theory of acoustics; the step of computing an impulse response computes said impulse response by convoluting said echotime pattern for each frequency band by a bandpass filter to compute a response for each frequency band, and by computing a composite response to compute said impulse response; and the step of generating reproduced sound data generates said reproduced sound data by receiving a dry source of the sound source, and by convoluting said impulse response and a head transfer function corresponding to the detected listener's location on said dry source of the sound source.

[0025] In the step of detecting data related to the listener's location, data related to dynamic changes of the listener's position and the direction of his head may be detected as the location data.

[0026] In the step of detecting data related to the listener's location, data related to directional changes of the listener's head may be detected as the location data.

[0027] The method may further comprise the step of dividing walls of the space into a set of sections; setting analysis conditions including the shape of the space, and physical data of the walls, the set of sections, a virtual sound source and a virtual sound receiving point; wherein the step of calculating said echotime pattern may include the steps of computing time series data related to arrival volume of a sound emanated from said sound source to said sound receiving point for each incident direction and frequency band; and computing said echotime pattern by interpolating certain data to said time series data.

[0028] The step of computing time series data may include the steps of computing data related to the arrival volume of the sound emanated directly from said sound source to said sound receiving point; computing data related to the arrival volume of the sound arriving at said sound receiving point by reflecting for up to a predetermined number of times by using a sound absorption coefficient and form factors of each section included in said physical data of the walls; computing data related to the arrival volume of the sound arriving at said sound receiving point by reflecting for more than the predetermined number of times by using the sound absorption coefficient and the form factors of each section included in said physical data of the walls; and computing time series data by dividing said arrival volume data into a set of groups by a certain time interval.

[0029] In the step of computing data related to the arrival volume of the sound arriving at said sound receiving point by reflecting for more than a predetermined number of times, a new virtual sound source may be set by integrating said arrival volume data of each group, and the arrival volume of sounds emanated from said new virtual sound source to any of the sections may be repeatedly computed.

BRIEF DESCRIPTION OF THE DRAWINGS

[0030] These and other objects, advantages and features of the invention will become apparent from the following description thereof taken in conjugation with the accompanying drawings which illustrate specific embodiments of the invention. In the drawings:

Fig. 1 is a block diagram of the sound environment simulator of the present invention;

Fig. 2 is a block diagram of the sound environment simulator in accordance with the first embodiment;

Fig. 3 is a flowchart detailing the operation of the sound environment analyzing unit 1;

Fig. 4 is a flowchart detailing the sub-routine of the step for computing the echotime pattern in Fig. 3;

Fig. 5 is an illustration depicting the model space used by the echotime pattern computing unit 15;

Fig. 6 is another illustration depicting the model space used by the echotime pattern computing unit 15;

Fig. 7 (a) is a graph of the time series data of the actual arrival volume at one of the divided section;

Fig. 7(b) is another graph of the time series data of the arrival volume divided into a set of groups at one of the divided section;

Fig. 8(a) is a graph of the time series data of the actual arrival volume at the sound receiving point;

Fig. 8(b) is a graph of the time series data of the arrival volume divided into a set of groups at the sound receiving point;

Fig. 8(c) is a graph of the time series data of the arrival volume interpolated with a certain data (echotime patter) at the sound receiving point;

Fig. 9 is a block diagram detailing the construction of the sound field reproducing unit 2 and output unit 3;

Fig. 10 is a block diagram detailing the construction of the impulse response computing unit 21;

Fig. 11 is a flowchart detailing the operation of the impulse response computing unit 21;

Fig. 12 is a flowchart detailing the operation of the sound field reproducing unit 2 in Fig. 9:

Fig. 13 is a block diagram of the sound environment simulator in accordance with the second embodi-

ment; and

Fig. 14 is a block diagram of the sound environment simulator in accordance with the third embodiment.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

[First Embodiment]

[0031] A sound environment simulator in accordance with the first embodiment reproduces a realistic, stereophonic sound by computing data related to acoustic characteristics in a virtual space created as a model environment, so that a listener can experience a simulation of a sound field.

[0032] The sound environment simulator, as is shown in Fig. 1, comprises a sound environment analyzing unit 1, a sound field reproducing unit 2, and an output unit 3. The sound environment analyzing unit 1 analyzes the acoustic characteristics in the virtual space by means of computer simulation to compute a echotime pattern. The sound field reproducing unit 2 computes data related to an impulse response from the echotime pattern to transduce non-reverberant portion of a desired sound into a consecutive sound signal on real time. The output unit 3 includes a headphone or the VR equipment and transduces the consecutive sound signal into a sound.

[0033] The above-described sound environment simulator is designed to simulate changes in the sound occurring as the listener stands up, sits down, walks around in a room, or moves his head. More precisely, the sound environment analyzing unit 1 computes the echotime patterns for each location, incident direction, and frequency of a sound source. The sound field reproducing unit 2 computes the impulse response from the echotime patterns, and adds further processing to produce a stereophonic sound signal, which is transduced into a sound by the output unit 3.

[0034] Fig. 2 provides a detailed construction of the sound environment analyzing unit 1. It comprises a room data input unit 11, a sound source data input unit 12, a sound receiving point data input unit 13, a surface dividing unit 14, an echotime pattern computing unit 15 including a time series data computing unit 16 and a data transducing unit 17, and a memory 18.

[0035] The sound environment analyzing unit 1 computes the echotime patterns for each location in the virtual space, each incident direction of the sounds incident to the sound receiving point, and each sound source including a plurality of predetermined frequency bands.

[0036] The flowchart in Fig. 3 details the above-described operation.

[0037] In Step 100, data related to the virtual space such as the shape, size and a sound absorption coefficient of walls are inputted into the room data input unit 11. Subsequently, data related to the location of the sound receiving point(location of the listener's head), and data related to incident direction of the sound to the sound receiving point are inputted at the sound receiving point data input unit 13 in Steps 110 and 120, respectively. Further, data related to the sound source such as the frequency and volume thereof are inputted at the sound source data input unit 12 in Step 130.

[0038] Then, in Step 140, the surface dividing unit 14 divides solid surfaces, or namely the walls, of the virtual space into a set of sections, and numbers them serially. Each section is a rectangle whose side is shorter than the wave length of the sounds emanated from the sound source. It also sets the sound absorption coefficient for each section from the sound absorption coefficient inputted in Step 100, and the sound receiving point. The sound receiving point takes a form of, for example, a dodecahedron to approximate the listener's head.

[0039] An example of the virtual space created in this way is illustrated as a model space 10 in Fig. 5. In the drawing, one wall is divided into approximately 400 sections, and the sound source is set at 500 Hz, whereby the short side of each section is set to 45cm. The sound absorption coefficient α is added to each section; the sound absorption coefficient αi is added to the section i, and αj to the section j. The frequency of the sound source is set per 1/8 octave band; for the sound absorption coefficient data of the walls are practically computed per octave band.

[0040] Once the model space 10 is created, the echotime pattern calculator 15 computes the echotime pattern in Step 200, which is detailed by the flowchart in Fig. 4.

[0041] In Step 210, the time series data computing unit 16 computes form factors. These form factors are calculated by a radiant heat ray tracing technique, which is often used in analyzing radiation in the thermal environment or lighting. Japanese laid-open patent application No. H5-52383, Hattori et al teaches the calculation method in detail. In this step, the form factors are calculated according to this calculation method with respect to the relations between: the sound source and each section; one section and another section; and each section and the sound receiving point, which are respectively referred to as Fsc, Fcc and Fcr.

[0042] The calculation of reflected sound with the form factors further adds reflection components having directivity and those of mirror reflection besides perfect reflection components to form factors as is with the lighting. Thus, more realistic reflections can be simulated by using this calculation. Also, this calculation is virtually a must when only an initial reflection by the mirror reflection at a relatively large divided section of a huge space such as a concert hall is roughly estimated, or when the reflection has the directivity due to the inclined walls.

[0043] More precisely, each form factor is calculated in the following way:

1) Form Factor Fsc (Sound Source → Section)
Suppose an arbitrary positioned virtual sound

source S evenly emanates a plurality of sound rays (e.g. 100,000 rays) in all directions. Then, specific sections at which each sound ray arrives are detected. For example, with the model space 10 in Fig. 5, a sound ray SRsk arrives at a section k, and sound rays $SRsi_1$ and $SRsi_2$ arrive at a section i. Accordingly, the time series data computing unit 16 computes the arrival volume P1 at each section to compute P1 : Ps, P1's ratio to the volume of the sound source Ps, which is referred to as the form factor Fsc. In this way, the form factors Fsc's for all the sections are calculated.

2) Form Factor Fcc (Section → Section)

For factor Fcc is calculated in the same manner. In this case, the arrival volume to one of the sections is considered to be a new sound source.

3) Form Factor Fcr (Section → Sound Receiving Point)

Form factor Fcr is calculated in the same manner.

[0044] Once these form factors are calculated, the time series data computing unit 16 computes time series data related to the arrival volume. Note that the sound receiving point R receives two types of sounds : one is a direct sound from the sound source, and the other is a reflected sound from one of the sections. Thus, the time series data computing unit 16 computes the time series data of the direct sound first, thence, those of the reflected sound. For example, with the model space 10, the sound emanated along with the sound ray SRsr is the direct sound, and the one with the sound ray SRsk is the indirect sound.

[0045] Back to the flowchart, the time series data computing unit 16 computes the arrival volume of the direct sound in Step 220. The arrival volume denoted as Pr is computed with the distance from the sound source to the sound receiving point R according to the inverse square law, while at the same time, the arrival time denoted as TR is computed by dividing the distance by the speed of sound.

[0046] Next in the Step 230, the time series data computing unit 16 computes the arrival volume of the reflected sound. The amount of this computation increases a geometric series-fold, because the number of reflected sound rays are multiplied by the number of the sections each time they reflect. Thus, a memory must have quite a large capacity to store these myriad computation results. To avoid this problem, the volume of the reflected sound is computed in two ways depending on the number of reflections, or largeness of the orders. In this embodiment, the reflections up to the third order are referred to as lower order reflections, and the reflections beyond the third order reflections as higher order reflections.

[0047] To begin with, a method for computing the arrival volume of the sound of the lower order refections to the sound receiving point R will be explained with referring to Fig. 5.

(The First Order Reflections)

[0048] Let the sound rays SRsk and SRkr be examples. The sound along with the sound rays are emanated from the sound source S and incident to the sound receiving point R by being reflected once at the section k. The volume thereof denoted as Pr_2 is computed with the form factors Fsc_1 and Fc_1r, and absorption coefficient αk according to :

$$Pr_2 = Fsc_1 \times Fc_1r \times (1 - \alpha k) \times Ps$$

[0049] Subsequently, like with the direct sound, the distances between the sound source S and the section k, the section k to the sound receiving point R are added and divided by the speed of sound to compute the arrival time T_2.

[0050] The above calculation is applied to all of the sounds of the first order reflections.

(The Second Order Reflections)

[0051] Let the sound rays $SRsi_2$, SRij and SRjr be examples. The sound along with these sound rays are reflected at the sections i and j before it arrives at the sound receiving point R. The arrival volume denoted as Pr_2 is computed with the form factors Fsc, Fcc_2 and Fc_2r, and absorption coefficients αi and αk according to :

$$Pr_2 = Fsc \times Fcc_2 \times Fc_2r \times (1 - \alpha i) \times (1 - \alpha k) \times Ps$$

[0052] Also, the distances between the sound source S and the section i, the sections i and the section j, and the section j to the sound receiving point R are added and divided by the speed of sound to compute the arrival time T_3.

[0053] The above calculation is applied to all of the sounds of the second order reflections.

(The Third Order Reflections)

[0054] The volume of the sound of all of the third order reflections is computed in the same manner described as above.

(The Higher Order Reflections)

[0055] Next, a method for computing the arrival volume of the sound of the higher order refections to the sound receiving point R will be explained with referring to Fig. 6.

[0056] Let the section n be an example, then the time series data of the third order reflections, SRjn and SRmn as is shown in Fig. 7(a), have been computed at the end

of the computation for the third order reflections.

[0057] These time series data are divided into a set of groups by a certain time interval Δt, for example, 0.1 Ms. The volume of each group is added to find a sum of the time series data within each group, and the resulting data, referred to as the time series data of the sum, is shown in Fig. 7(b). Then, the section having the time series data of the sum is positioned as a new virtual sound source S'. In practice, at the new sound source S', N sound rays arriving at the concerned section during Δt are composited into a single sound ray, so that it will have a volume equal to the sum of the volume of the N sound rays'.

[0058] Back to the flowchart again, in Step 250, as is with the sound of the lower order reflections, the arrival volume along with the sound rays to either the sound receiving point R or another section are computed. This computation is repeated until the arrival volume of all the sections becomes 1/10000 of the entire emanated volume Ps or less.

[0059] During the above-described computation, the sound receiving point R receives a large amount of the direct and indirect sounds as is shown in the graph in Fig. 8(a). The arrival sound rays are divided into a set of groups as is with the higher order reflections, for example by a certain time interval, then the number thereof is counted, and the arrival volume of each sound ray are summed up if they are plural.

[0060] When the arrival volume of all the sections becomes lower than the predetermined level in Step 250, the further computation is not proceeded. Instead, such a petty arrival amount are summed up in each section, and these sums are further added to the time series data in Step 260.

[0061] As a result, the time series data of the arrival volume at the sound receiving point R, as is shown in Fig. 8(b), is computed.

[0062] Accordingly, the data transducing unit 17 computes a mean arrival volume in Step 270 to interpolates data to generate the echotime pattern from the time series data of the sum in the following step 280.

[0063] More precisely, the data transducing unit 17 computes a mean arrival volume data as is shown in Fig. 8(c) from the time series data of the sum. To compute the mean arrival volume data, the sum of the arrival volumes at the receiving point R within a certain group is divided by the number of the arrival sound rays. Assuming that a composite sound ray SRr consisting of N (=3) sound rays arrives to the sound receiving point R with a arrival volume Pm during the third group(0.2-0.3 Ms), then the mean volume data SRa is computed according to :

$$SRa = Pm/N$$

[0064] Subsequently, to interpolate data to the mean volume data SRa thus calculated, the data transducing unit 17 applies the mean volume data of each group to the intermediate point of the corresponding group. Then, as is shown in Fig. 8(c), the tops of each bar in the graph representing the mean volume data of the groups are linked, so that an approximate line will be formed. By using this approximate line, the data converting unit 17 interpolates as many new lines as the actual arrival N sound rays in the corresponding group. As a result of this data interpolation, the echotime pattern of an approximate line as is shown in Fig. 8(c) is formed. This interpolation is not limited to the approximation by a line; a curve may be used instead of a line, or an addition of a weight corresponding to the arrival time to each sound ray also will do.

[0065] The optimal effects are obtained when Δt is set to a corresponding value to a sampling frequency of the audio equipments such as compact disc player; for example, the optimal Δt for a sampling frequency of 44.1 KHz is 22.7 Ms.

[0066] The echotime pattern thus computed is stored into the memory 18, and read out for further processing at the sound field reproducing unit 2.

[0067] Fig. 9 provides a detailed construction of these two components. The sound field reproducing unit 2 comprises an impulse response computing unit 21 for reading out the echotime pattern from the memory 18 to compute an impulse response, a location data input unit 22 for reading out the data of the location of the listener within the model space; the location data are detected by a location data detecting unit 31 such as the VR equipment in the output unit 3, a head transfer function storing unit 23 for storing a predetermined head transfer function, a dry source input unit 24 for inputting data related to a dry source of a sound(the sound recorded in an anechoic room) to be reproduced, a reproduced sound generating unit 25 for generating a consecutive sound signal by convoluting data related to the impulse response and head transfer function on the data related to the dry source with a plurality of digital signal processors(DSPs).

[0068] As is shown in Fig. 10, the impulse response computing unit 21 includes a echotime pattern input unit 41 for inputting data related to the echotime pattern for each location in the model space, each incident direction, and each frequency band, a convolution computing unit 42 for convoluting the echotime pattern of each frequency band by a bandpass filter, an impulse response computing unit 43 for computing the impulse response by compositing responses from all the frequency bands, an impulse response output unit 44 for outputting the impulse response for each location and incident direction, and a memory 45 for storing data related to the impulse response.

[0069] The output unit 3 includes a sound output instrument 32 such as a headphone, over which a sound based on the consecutive signal from the reproduced sound generating unit 25 is reproduced, so that the listener can hear it.

[0070] In practice, a RB2(Reality Built for 2, a trade name) system or an Eyephone system of VPL Research Inc, and POLHEMUS are used for the location data detecting unit 31. In the Eyephone system, a sensor placed on top of the listener's head detects his location and his head's direction to output the related data to the sound field reproducing unit 2 on real time.

[0071] The above described sound field reproducing unit 2 and output unit 3 operate in the following way.

[0072] The impulse response computing unit 21 computes the impulse response for each location and incident direction (listener's head direction) from the echotime pattern.

[0073] The flowchart in Fig. 11 details how the impulse response is generated from the echotime pattern.

[0074] In Step 300, the impulse response computing unit 21 reads out the echotime pattern sequentially from the memory 18, and in Step 320, it computes a response for each frequency band by convoluting each frequency by a bandpass filter. Consequently, the impulse data related to all the locations in the model space and incident direction are computed and stored in the memory 45.

[0075] Once the impulse data are computed, the sound field reproducing unit 2 shifts to a sound reproducing process, which is detailed by the flowchart in Fig. 12.

[0076] In Step 400, the sound field reproducing unit 2 reads out the location data from the location data input unit 22 on real time. Then, the impulse response input unit 41 reads out the impulse response data corresponding to the location data from the memory 45 in Step 410. Further, it reads out the head transfer function data from the head transfer function storing unit 43 corresponding to the location data as to the listener's head direction in Step 420. As previously mentioned, the head transfer function is a parameter that represents the change in an incident sound in accordance with the shape of the listener's head and a location of his ears, and stored in the head transfer function storing unit 43 in advance.

[0077] Subsequently, in Step 430, the reproduced sound generating unit 25 receives the dry source of the sound from the dry source input unit 24, and convolutes the impulse response data on head transfer function data with the DSPs to generate a consecutive sound signal, which is outputted to the sound output unit 52 on real time and transduced into a sound in Step 440.

[0078] As a result, the listener can hear a realistic, stereophonic sound as if he were in the model space.

[Second Embodiment]

[0079] The sound environment simulator in accordance with the second embodiment is detailed in Fig. 13. The essential construction thereof is identical with the simulator of the first embodiment except that the sound environment analyzing unit 1 includes the echotime pattern computing unit 15 and impulse response computing unit 21; that the impulse response data for every location are computed with a computing server and stored into a memory in advance; and that the sound field reproducing unit 2 reads out the impulse response data and a head transfer function corresponding the location data from the memory to reproduce the stereophonic data from the dry source on real time. Otherwise, like components are labeled with like reference numerals with respect to the first embodiment, and the description of these component is not repeated.

[Third Embodiment]

[0080] The sound environment simulator in accordance with the third embodiment is detailed in Fig. 14. The essential construction thereof is identical with the simulator of the first embodiment except that the sound environment analyzing unit 1 includes the head transfer storing unit 23 and a composite response data computing unit 26. The composite response data computing unit 26 convolutes the head transfer function data and impulse response data to compute the composite response data in advance, and accordingly, the sound reproducing unit 24 convolutes the composite response data for each head direction into the dry source on real time with the DSPs. Otherwise, like components are labeled with like reference numerals with respect to the first embodiment, and the description of these component is not repeated.

[0081] With the simulator constructed as above, the number of the DSPs can be decreased as the number of incident directions increases. At the same time, less the data amount to be transferred to the DSPs becomes, faster the sound environment simulator can correspond to the dynamic changes of the listener.

[0082] As has been explained, computing the time series data of the arrival volume enables a precise prediction of the reverberant time. It also allows an accurate computation of the frequency characteristics of a reverberant sound. Thus, one or a few DPSs can also serve as a delaying circuit when the time length of the reverberant sound and the frequency characteristics are analyzed and approximated. As a result, the realistic sound field can be reproduced by using less DSPs without impairing the listener's feeling of his presence in the model space.

[0083] The reflected sounds are divided into the lower order reflections and higher order reflections and analyzed by the respective computation methods to enhance the work efficiency of the memory. However, the algorithm used for the higher order reflections can be used to all of the reflected sounds.

[0084] In all the three embodiments, the sound environment simulator is designed to simulate the sound environment which adequately corresponds to the changes occurring when the listener sits, stands up, walks around or moves his head. However, it may be designed to apply to a specific situation alone.

[0085] If the sound environment simulator is to be applied to a case where the listener do not change his position, the sound receiving point R can be fixed to a single point, which obviates the location data detecting unit 31 and location data input unit 22, contributing to simplifying the construction.

[0086] If the sound environment simulator is to be applied to a case when the listener moves his head in the same position, the sound receiving point R can be fixed to a single point as well, and the location data detecting unit 31 is only to detect the directional change of the listener's head. Hence the amount of data used for computation decreases, which results in enhancing the overall operation speed.

[0087] If the sound environment simulator is to apply to a case when the listener changes his position as he walks around, the height from the floor level to the listener's ears have a constant value. Therefore, the three dimensional location data can be replaced with two dimensional data. Hence, the overall operation speed can be enhanced for the same reason described in the previous paragraph.

[0088] With the sound environment simulator of the present invention, there may be a plurality of virtual sound sources, and even when they change their locations, the realistic, stereophonic sound can be reproduced as well; for it reproduces an adequate sound field by adding the impulse response data for each incident direction accordingly.

[0089] The confined space was created as the virtual space in all the embodiments; however, it can be an open space if equations for the open space are used to compute the echotime patterns. Also, the present invention enables the computation of the arrival volume of the sound of the higher order reflections as has been described. Further, it can transduce the echotime pattern into the impulse response data in accordance with the listener's arbitrary position and head direction. Consequently, the present invention reproduces the realistic sound field by means of the existing VR equipment, which improves efficiency in designing for an audio and living room.

[0090] Although the present invention has been fully described by way of example with reference to the accompanying drawings, it is to be noted that various changes and modification will be apparent to those skilled in the art. Therefore, unless otherwise such changes and modifications depart from the scope of the present invention, they should be construed as being included therein.

Claims

1. A sound environment simulator comprising:

    echotime pattern computing means (15) for computing an echotime pattern of a space;
    impulse response computing means (21) for computing an impulse response using said echotime pattern;
    reproduced sound generating means (25) for generating reproduced sound data using said impulse response; and
    output means (3) for outputting a sound by transducing said reproduced sound data into the sound;

    said simulator being characterised in that:

    said echotime pattern computing means (15) is adapted to compute said echotime pattern by a geometric method based on the theory of acoustics with input data as to conditions of said space;
    said impulse response computing means (21) is adapted to compute a response for each frequency band from said echotime pattern and to compute a composite response to generate said impulse response; and
    said reproduced sound generating means (25) is adapted to generate the reproduced sound data by convolution of said impulse response on non-reverberate sound source data.

2. The simulator of claim 1, wherein said simulator further comprises analysis data setting means (11-14) for setting analysis data by receiving analysis condition data including a shape and physical data of walls of said space, dividing said walls into a set of certain sections, and setting a virtual sound source and a virtual sound receiving point at respective arbitrary positions within said space; and

    said echotime pattern means (15) is adapted to receive said setting analysis data and includes:

    time series data computing means (16) for computing, based on said setting analysis data, time series data related to arrival volume of a sound emanated from said virtual sound source to said virtual sound receiving point; and
    data transducing means (17) for transducing said time series data related to the arrival volume into said echotime pattern by interpolating certain data.

3. The simulator of claim 2 further comprising:

    dry source input means (24) for inputting said non-reverberate sound source data; and
    head transfer function storing means (23) for storing data related to a head transfer function;
    wherein said reproduced sound generating means (25) is adapted to generate the reproduced sound data by convoluting said impulse response received from said impulse response

computing means (21) and said head transfer function received from said head transfer function storing means (23) on said non-reverberate sound source data received from said dry source input means (24).

4. The simulator of claim 3, further comprising:

location data detecting means (31) for detecting data related to at least one of a listener's head location and a listener's head direction;
wherein said reproduced sound generating means (25) is adapted to extract said impulse response and said head transfer function corresponding to said detected location data to convolute them on said non-reverberate sound source data.

5. The simulator of claim 4, wherein said location data detecting means (31) is adapted to detect time varying data.

6. The simulator of claim 4, wherein said output means (32) comprises a headphone.

7. The simulator of claim 1, further comprising:

head transfer function storing means (23) for storing data related to a head transfer function; and
composite response generating means (26) for computing a composite response for each direction of a listener's head by convolution of said impulse response and said head transfer function,
wherein said reproduced sound generating means (25) is adapted to generate reproduced sound data by convolution of said composite response on the non-reverberate sound source data.

8. The simulator of claim 2, wherein said time series data computing means (16) is adapted to compute form factors to compute the arrival volume of the sound to the sound receiving point by using said form factors and a sound absorption coefficient of each section included in said physical data of the walls.

9. A method of analysing a sound space, comprising the steps of:

computing an echotime pattern of a space;
computing an impulse response using said echotime pattern;
generating reproduced sound data related to a reproduced sound based on said impulse response; and
outputting a sound by transducing said reproduced sound data in the sound;

said method being characterised by the step of:

detecting data related to a listener's location;
wherein the step of computing an echotime pattern computes said echotime pattern for a virtual sound source placed arbitrarily in the space by a geometric method based on the theory of acoustics;
the step of computing an impulse response computes said impulse response by convoluting said echotime pattern for each frequency band by a bandpass filter to compute a response for each frequency band, and by computing a composite response to compute said impulse response; and
the step of generating reproduced sound data generates said reproduced sound data by receiving a dry source of the sound source, and by convoluting said impulse response and a head transfer function corresponding to the detected listener's location on said dry source of the sound source.

10. The method of claim 9, wherein in the step of detecting data related to the listener's location, data related to dynamic changes of the listener's position and the direction of his head are detected as the location data.

11. The method of claim 9, wherein in the step of detecting data related to the listener's location, data related to directional changes of the listener's head are detected as the location data.

12. The method of claim 9, further comprising the step of:

dividing walls of the space into a set of sections;
setting analysis conditions including the shape of the space, and physical data of the walls, the set of sections, a virtual sound source and a virtual sound receiving point;
wherein the step of calculating said echotime pattern includes the steps of:
computing time series data related to arrival volume of a sound emanated from said sound source to said sound receiving point for each incident direction and frequency band; and
computing said echotime pattern by interpolating certain data to said time series data.

13. The method of claim 12, wherein the step of computing time series data includes the steps of:

computing data related to the arrival volume of the sound emanated directly from said sound source to said sound receiving point;
computing data related to the arrival volume of the sound arriving at said sound receiving point by reflecting for up to a predetermined number of times by using a sound absorption coefficient and form factors of each section included in said physical data of the walls;
computing data related to the arrival volume of the sound arriving at said sound receiving point by reflecting for more than the predetermined number of times by using the sound absorption coefficient and the form factors of each section included in said physical data of the walls; and
computing time series data by dividing said arrival volume data into a set of groups by a certain time interval.

14. The method of claim 13, wherein in the step of computing data related to the arrival volume of the sound arriving at said sound receiving point by reflecting for more than a predetermined number of times, a new virtual sound source is set by integrating said arrival volume data of each group, and the arrival volume of sounds emanated from said new virtual sound source to any of the sections are repeatedly computed.

Patentansprüche

1. Schallfeldumgebungssimulator, der aufweist:

eine Echolaufzeit-Muster-Berechnungseinrichtung (15) zum Berechnen eines Echolaufzeit-Musters eines Raums;

eine Impulsansprechverhalten-Berechnungseinrichtung (21) zum Berechnen eines Impulsansprechverhaltens unter Verwendung des Echolaufzeit-Musters;

eine Schallwiedergabe-Erzeugungseinrichtung (25) zum Erzeugen reproduzierter Schalldaten unter Verwendung des Impulsansprechens; und

eine Ausgabeeinrichtung (3) zum Ausgeben von Schall durch Umwandeln der reproduzierten Schalldaten in den Schall;

wobei der Simulator dadurch gekennzeichnet ist, daß die Echolaufzeit-Muster-Berechnungseinrichtung (15) so angepaßt ist, um das Echolaufzeit-Muster durch ein geometrisches Verfahren basierend auf der Akkustik-Theorie mit Eingabedaten als Bedingungen für den Raum zu berechnen;

daß die Impulsansprechverhalten-Berechnungseinrichtung (21) dazu angepaßt ist, ein Ansprechen auf jedes Frequenzband auf dem Echolaufzeit-Muster zu berechnen und ein Komposit-Ansprechverhalten zu berechnen, um das Impulsansprechverhalten zu erzeugen; und

wobei die Schallwiedergabe-Erzeugungseinrichtung (25) so angepaßt ist, um die reproduzierten Schalldaten durch Konvolution des Impulsansprechverhaltens auf den nicht widerhallenden Schallquellendaten zu erzeugen.

2. Simulator nach Anspruch 1, wobei der Simulator weiterhin eine Daten-Analyseeinstelleinrichtung (11-14) zum Einstellen von Analysedaten durch Aufnehmen von Analysezustandsdaten, die Form- und physikalische Daten der Wände des Raums umfassen, Aufteilen der Wände in einen Satz von bestimmten Abschnitten und Einstellen einer virtuellen Schallquelle und eines virtuellen Schallaufnahmepunkts an jeweiligen, wahlweisen Positionen innerhalb des Raums aufweist; und
wobei die Echolaufzeit-Muster-Einrichtung (15) so angepaßt ist, um die Einstellanalysedaten aufzunehmen, und umfaßt:

eine Zeitfolgedaten-Berechnungseinrichtung (16) zum Berechnen, basierend auf den Einstellanalysedaten, von Zeiffolgedaten, die sich auf ein Ankunftsvolumen von Schall beziehen, der von der virtuellen Schallquelle zu dem virtuellen Schallaufnahmepunkt abgestrahlt wird; und

eine Datenumwandlungseinrichtung (17) zum Umwandeln der Zeiffolgedaten, die sich auf das Ankunftsvolumen beziehen, in das Echolaufzeit-Muster durch Interpolieren bestimmter Daten.

3. Simulator nach Anspruch 2, der weiterhin aufweist:

eine Trockenquelleneingabeeinrichtung (24) zum Eingeben der nicht widerhallenden Schallquellendaten; und

eine Kopfübertragungsfunktionsspeichereinrichtung (23) zum Speichern von Daten, die sich auf eine Kopfübertragungsfunktion beziehen;

wobei die Schallwiedergabe-Erzeugungseinrichtung (25) so angepaßt ist, um die reproduzierten Schalldaten durch Konvolution des Im-

pulsansprechens, das von der Impulsansprechverhalten-Berechnungseinrichtung (21) erhalten ist, und der Kopfübertragungsfunktion, die von der Kopfübertragungsfunktionsspeichereinrichtung (23) erhalten ist, auf den nicht widerhallenden Schalldaten, die von der Trockenquelleneingabeeinrichtung (24) aufgenommen sind, zu erzeugen.

4. Simulator nach Anspruch 3, der weiterhin aufweist:

eine Lagedatenerfassungseinrichtung (31) zum Erfassen von Daten, die sich auf mindestens eine Kopfstelle eines Hörers und eine Kopfrichtung eines Hörers beziehen;

wobei die Schallwiedergabe-Erzeugungseinrichtung (25) dazu angepaßt ist, das Impulsansprechen und die Kopfübertragungsfunktion entsprechend den erfaßten Lagedaten zu extrahieren, um sie auf die nicht widerhallenden Schalldaten zu konvolutieren.

5. Simulator nach Anspruch 4, wobei die Lagedatenerfassungseinrichtung (31) so angepaßt ist, um Daten zu erfassen, die sich über die Zeit variieren.

6. Simulator nach Anspruch 4, wobei die Ausgabeeinrichtung (32) einen Kopfhörer aufweist.

7. Simulator nach Anspruch 1, der weiterhin aufweist:

eine Kopfübertragungsfunktionsspeichereinrichtung (23) zum Speichern von Daten, die sich auf eine Kopfübertragungsfunktion beziehen; und

eine Komposit-Ansprechverhalten-Erzeugungseinrichtung (26) zum Berechnen eines Komposit-Ansprechverhaltens für jede Richtung des Kopfes eines Hörers durch Konvolution des Impulsansprechverhaltens und der Kopfübertragungsfunktion,

wobei die Schallwiedergabe-Erzeugungseinrichtung (25) dazu angepaßt ist, reproduzierte Schalldaten durch Konvolution des Komposit-Ansprechverhaltens auf den nicht widerhallenden Schallquellendaten zu erzeugen.

8. Simulator nach Anspruch 2, wobei die Zeiffolgedaten-Berechnungseinrichtung (16) so angepaßt ist, um Formfaktoren zu berechnen, um das Ankunftsvolumen des Schalls an dem Schallaufnahmepunkt unter Verwendung der Formfaktoren und eines Schallabsorptionskoeffizienten jedes Abschnitts. der in den physikalischen Daten der Wände enthalten ist, zu berechnen.

9. Verfahren zum Analysieren eines Schallraums, das die Schritte aufweist:

Berechnen eines Echolaufzeit-Musters eines Raums;

Berechnen eines Impulsansprechverhaltens unter Verwendung des Echolaufzeit-Musters;

Erzeugen von Schallwiedergabedaten, die zu einem reproduzierten Schall basierend auf dem Impulsansprechverhalten in Bezug gesetzt sind; und

Ausgeben von Schall durch Wandeln der reproduzierten Schalldaten in den Schall;

wobei das Verfahren gekennzeichnet ist durch die Schritte:

Erfassen von Daten, die sich auf die Stelle eines Hörers beziehen;

wobei der Schritt eines Berechnens eines Echolaufzeit-Musters das Echolaufzeit-Muster aus einer virtuellen Schallquelle, die wahlweise in dem Raum plaziert ist, durch ein geometrisches Verfahren, basierend auf der Akustik-Theorie, berechnet;

wobei der Schritt eines Berechnens eines Impulsansprechens das Impulsansprechen durch Konvolutieren des Echolaufzeit-Musters für jedes Frequenzband durch einen Bandpaßfilter berechnet, um ein Ansprechverhalten für jedes Frequenzband zu berechnen, und durch Berechnen eines Komposit-Ansprechverhaltens, um das Impulsansprechverhalten zu berechnen; und wobei

der Schritt der Erzeugung reproduzierter Schalldaten die reproduzierten Schalldaten durch Aufnehmen einer Trockenquelle der Schallquelle erzeugt, und durch Konvolutieren des Impulsansprechverhaltens und einer Kopfübertragungsfunktion entsprechend der erfaßten Stelle des Hörers auf der Trockenquelle der Schallquelle erzeugt.

10. Verfahren nach Anspruch 9, wobei in dem Schritt eines Erfassens von Daten, die sich auf die Stelle des Hörers beziehen, Daten, die sich auf dynamische Änderungen in der Position des Hörers und der Richtung seines Kopfs beziehen, als die Lagedaten erfaßt werden.

11. Verfahren nach Anspruch 9, wobei in dem Schritt eines Erfassens von Daten, die sich auf die Stelle

des Hörers beziehen, Daten, die sich auf richtungsmäßige Änderungen des Kopfs des Hörers beziehen, als die Lagedaten erfaßt werden.

12. Verfahren nach Anspruch 9, das weiterhin den Schritt aufweist:

Unterteilen von Wänden des Raums in einen Satz von Abschnitten;

Einstellen von Analysebedingungen, die die Form des Raums, und physikalische Daten der Wände, den Satz von Abschnitten, eine virtuelle Schallquelle und einen virtuellen Schallaufnahmepunkt umfassen;

wobei der Schritt eines Berechnens des Echolaufzeit-Musters die Schritte umfaßt:

Berechnen von Zeiffolgedaten, die sich auf ein Ankunftsvolumen eines Schalls beziehen, der von der Schallquelle zu dem Schallaufnahmepunkt für jede einfallende Richtung und jedes Frequenzband abgegeben ist; und

Berechnen des Echolaufzeit-Musters durch Interpolieren bestimmter Daten zu den Zeiffolgedaten.

13. Verfahren nach Anspruch 12, wobei der Schritt eines Berechnens der Zeiffolgedaten die Schritte umfaßt:

Berechnen von Daten, die sich auf das Ankunftsvolumen des Schalls beziehen, der direkt von der Schallquelle abgegeben ist, zu dem Schallaufnahmepunkt;

Berechnen von Daten, die sich auf das Ankunftsvolumen des Schalls beziehen, der an dem Schallaufnahmepunkt ankommt, durch Reflektieren von bis zu einer vorbestimmten Anzahl von Malen unter Verwendung eines Schallabsorptionskoeffizienten und von Formfaktoren jedes Abschnitts, die in den physikalischen Daten der Wände umfaßt sind;

Berechnen von Daten, die sich auf das Ankunftsvolumen des Schalls beziehen, der an dem Schallaufnahmepunkt ankommt, durch Reflektieren mehr als die vorbestimmte Anzahl von Malen unter Verwendung des Schallabsorptionskoeffizienten und der Formfaktoren jedes Abschnitts, die in den physikalischen Daten der Wände umfaßt sind; und

Berechnen von Zeiffolgedaten durch Unterteilen der Ankunftsvolumendaten in einem Satz von Gruppen durch ein bestimmtes Zeitintervall.

14. Verfahren nach Anspruch 13, wobei in den Schritt eines Berechnens von Daten, die sich auf das Ankunftsvolumen des Schalls beziehen, der an dem Schallaufnahmepunkt durch Reflektieren mehr als eine vorbestimmte Anzahl von Malen ankommt, eine neue, virtuelle Schallquelle durch Integrieren der Ankunftsvolumendaten jeder Gruppe eingestellt wird, und wobei das Ankunftsvolumen des Schalls, der von der neuen, virtuellen Schallquelle zu irgendeinem der Abschnitte abgestrahlt wird, wiederholt berechnet wird.

Revendications

1. Simulateur d'environnement sonore comprenant :

des moyens de calcul de configuration de temps d'écho (15) pour calculer une configuration de temps d'écho d'un espace ;
des moyens de calcul de réponse impulsionnelle (21) pour calculer une réponse impulsionnelle en utilisant ladite configuration de temps d'écho ;
des moyens de génération de son reproduit (25) pour générer des données de son reproduit en utilisant ladite réponse impulsionnelle ; et
des moyens de sortie (3) pour sortir un son en convertissant lesdits données de son reproduit en son ;

ledit simulateur étant caractérisé en ce que :

lesdits moyens de calcul de configuration de temps d'écho (15) sont adaptés pour calculer ladite configuration de temps d'écho par un procédé géométrique basé sur la théorie de l'acoustique avec des données d'entrée quant aux conditions dudit espace ;
lesdits moyens de calcul de réponse impulsionnelle (21) sont adaptés pour calculer une réponse pour chaque bande de fréquences à partir de ladite configuration de temps d'écho et pour calculer une réponse composite pour générer ladite réponse impulsionnelle ; et
lesdits moyens de génération de son reproduit (25) sont adaptés pour générer les données de son reproduit par la convolution de ladite réponse impulsionnelle sur les données de source de son sans réverbération.

2. Simulateur selon la revendication 1, dans lequel ledit simulateur comprend, de plus, des moyens d'établissement de données d'analyse (11 à 14)

pour établir des données d'analyse en recevant les données de condition d'analyse comprenant une forme et des données physiques des parois dudit espace, pour diviser lesdites parois en un ensemble de certaines sections, et pour établir une source de son virtuelle et un point de réception du son virtuel aux positions arbitraires respectives dans ledit espace ; et lesdits moyens de configuration de temps d'écho (15) sont adaptés pour recevoir lesdites données d'analyse établies et comprennent :

des moyens de calcul de données sérielles dans le temps (16) pour calculer, en se basant sur lesdites données d'analyse établies, des données sérielles dans le temps en rapport avec le volume à l'arrivée d'un son provenant de ladite source de son virtuelle audit point de réception du son virtuel ; et
des moyens de conversion de données (17) pour convertir lesdites données sérielles dans le temps en rapport avec le volume à l'arrivée en ladite configuration de temps d'écho en interpolant certaines données.

3. Simulateur selon la revendication 2, comprenant, de plus :

des moyens d'entrée de source sèche (24) pour entrer lesdites données de source de son sans réverbération ; et des moyens de mémorisation de fonction de transfert de tête (23) pour mémoriser des données en rapport avec une fonction de transfert de tête ;
dans lequel lesdits moyens de génération de son reproduit (25) sont adaptés pour générer les données de son reproduit par la convolution de ladite réponse impulsionnelle reçue en provenance desdits moyens de calcul de réponse impulsionnelle (21) et de ladite fonction de transfert de tête reçue en provenance desdits moyens de mémorisation de fonction de transfert de tête (23) sur lesdites données de source de son sans réverbération reçues en provenance desdits moyens d'entrée de source sèche (24).

4. Simulateur selon la revendication 3, comprenant, de plus :

des moyens de détection de données d'emplacement (31) pour détecter des données en rapport avec au moins l'un de l'emplacement de la tête d'un auditeur et de la direction de la tête d'un auditeur ;
dans lequel lesdits moyens de génération de son reproduit (25) sont adaptés pour extraire ladite réponse impulsionnelle et ladite fonction de transfert de tête correspondant auxdites données d'emplacement détectées pour les convoluer sur lesdites données de source de son sans réverbération.

5. Simulateur selon la revendication 4, dans lequel lesdits moyens de détection de données d'emplacement (31) sont adaptés pour détecter des données variant dans le temps.

6. Simulateur selon la revendication 4, dans lequel lesdits moyens de sortie (32) comprennent un écouteur.

7. Simulateur selon la revendication 1, comprenant, de plus :

des moyens de mémorisation de fonction de transfert de tête (23) pour mémoriser des données en rapport avec une fonction de transfert de tête ; et
des moyens de génération de réponse composite (26) pour calculer une réponse composite pour chaque direction de la tête d'un auditeur par la convolution de ladite réponse impulsionnelle et de ladite fonction de transfert de tête,
dans lequel lesdits moyens de génération de son reproduit (25) sont adaptés pour générer des données de son reproduit par la convolution de ladite réponse composite sur les données de source de son sans réverbération.

8. Simulateur selon la revendication 2, dans lequel lesdits moyens de calcul de données sérielles dans le temps (16) sont adaptés pour calculer des facteurs de forme pour calculer le volume à l'arrivée du son au point de réception du son en utilisant lesdits facteurs de forme et un coefficient d'absorption du son de chaque section comprise dans lesdites données physiques des parois.

9. Procédé d'analyse d'un espace sonore, comprenant les étapes consistant à :

calculer une configuration de temps d'écho d'un espace ;
calculer une réponse impulsionnelle en utilisant ladite configuration de temps d'écho ;
générer des données de son reproduit en rapport avec un son reproduit sur la base de ladite réponse impulsionnelle ; et
sortir un son en convertissant lesdits données de son reproduit dans le son ;

ledit procédé étant caractérisé par l'étape consistant à :

détecter des données en rapport avec l'emplacement d'un auditeur ;

dans lequel l'étape consistant à calculer une configuration de temps d'écho calcule ladite configuration de temps d'écho pour une source de son virtuelle placée arbitrairement dans l'espace par un procédé géométrique basé sur la théorie de l'acoustique ;
l'étape consistant à calculer une réponse impulsionnelle calcule ladite réponse impulsionnelle par la convolution de ladite configuration de temps d'écho pour chaque bande de fréquences par un filtre passe-bande pour calculer une réponse pour chaque bande de fréquences, et par le calcul d'une réponse composite pour calculer ladite réponse impulsionnelle ; et
l'étape consistant à générer des données de son reproduit génère lesdits données de son reproduit par la réception d'une source sèche de la source de son, et par la convolution de ladite réponse impulsionnelle et d'une fonction de transfert de tête correspondant à l'emplacement détecté de l'auditeur sur ladite source sèche de la source de son.

10. Procédé selon la revendication 9, dans lequel, dans l'étape consistant à détecter des données en rapport avec l'emplacement de l'auditeur, les données en rapport avec les changements dynamiques de la position de l'auditeur et de la direction de sa tête sont détectées comme les données d'emplacement.

11. Procédé selon la revendication 9, dans lequel dans l'étape consistant à détecter des données en rapport avec l'emplacement de l'auditeur, les données en rapport avec les changements de direction de la tête de l'auditeur sont détectées comme les données d'emplacement.

12. Procédé selon la revendication 9, comprenant, de plus, l'étape consistant à :

diviser les parois de l'espace en un ensemble de sections ;
établir des conditions d'analyse comprenant la forme de l'espace, et les données physiques des parois, l'ensemble de sections, une source de son virtuelle et un point de réception du son virtuel ;
dans lequel l'étape consistant à calculer ladite configuration de temps d'écho comprend les étapes consistant à :
calculer des données sérielles dans le temps en rapport avec le volume à l'arrivée d'un son provenant de ladite source de son audit point de réception du son pour chaque direction incidente et chaque bande de fréquences ; et
calculer ladite configuration de temps d'écho en interpolant certaines données à partir desdites données sérielles dans le temps.

13. Procédé selon la revendication 12, dans lequel l'étape consistant à calculer des données sérielles dans le temps comprend les étapes consistant à :

calculer des données en rapport avec le volume à l'arrivée du son provenant directement de ladite source de son audit point de réception du son ;
calculer des données en rapport avec le volume à l'arrivée du son arrivant audit point de réception du son après un nombre prédéterminé de réflexions en utilisant un coefficient d'absorption du son et les facteurs de forme de chaque section comprise dans lesdites données physiques des parois ;
calculer des données en rapport avec le volume à l'arrivée du son arrivant audit point de réception du son après un nombre de réflexions supérieur au nombre prédéterminé en utilisant le coefficient d'absorption du son et les facteurs de forme de chaque section comprise dans lesdites données physiques des parois ; et
calculer des données sérielles dans le temps en divisant lesdites données de volume à l'arrivée en un ensemble de groupes par un certain intervalle de temps.

14. Procédé selon la revendication 13, dans lequel, dans l'étape consistant à calculer des données en rapport avec le volume à l'arrivée du son arrivant audit point de réception du son après un nombre de réflexions supérieur au nombre prédéterminé, une nouvelle source de son virtuelle est établie en intégrant lesdites données de volume à l'arrivée de chaque groupe, et les volumes à l'arrivée des sons provenant de ladite nouvelle source de son virtuelle à l'une quelconque des sections sont calculés à plusieurs reprises.

Fig.1

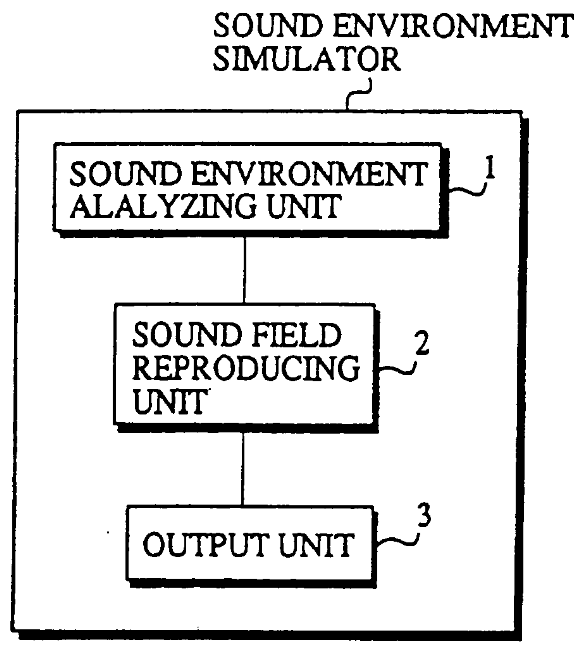

SOUND ENVIRONMENT SIMULATOR
SOUND ENVIRONMENT ALALYZING UNIT
1
SOUND FIELD REPRODUCING UNIT
2
OUTPUT UNIT
3

Fig.2

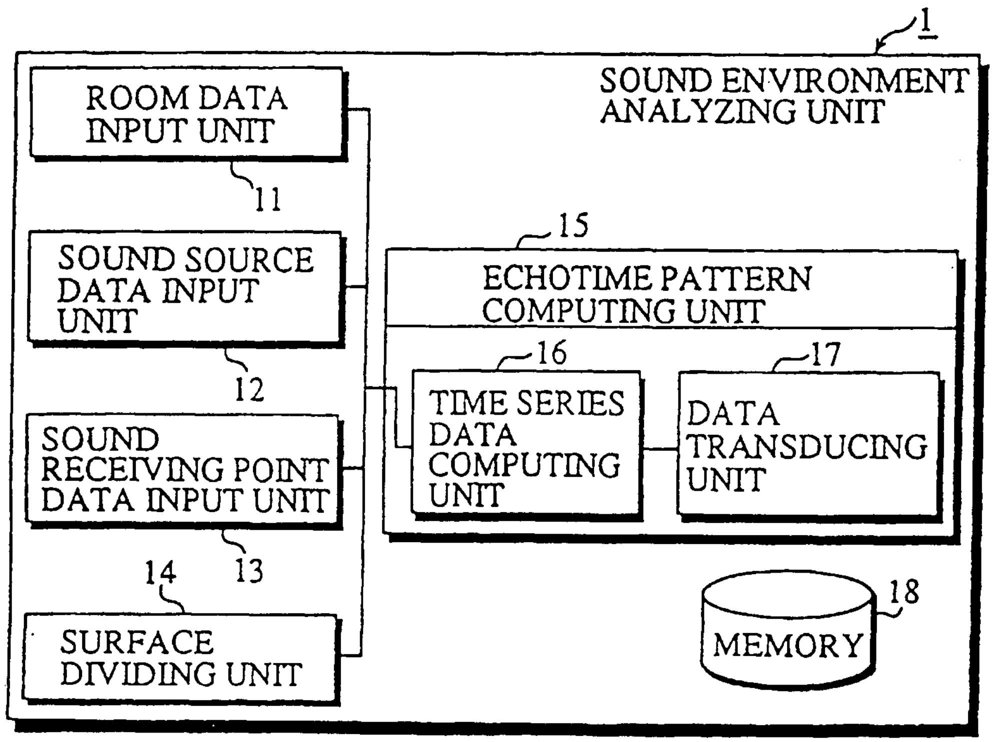

1
SOUND ENVIRONMENT ANALYZING UNIT
ROOM DATA INPUT UNIT
11
SOUND SOURCE DATA INPUT UNIT
12
SOUND RECEIVING POINT DATA INPUT UNIT
13
14
SURFACE DIVIDING UNIT
15
ECHOTIME PATTERN COMPUTING UNIT
16
TIME SERIES DATA COMPUTING UNIT
17
DATA TRANSDUCING UNIT
18
MEMORY

Fig.3

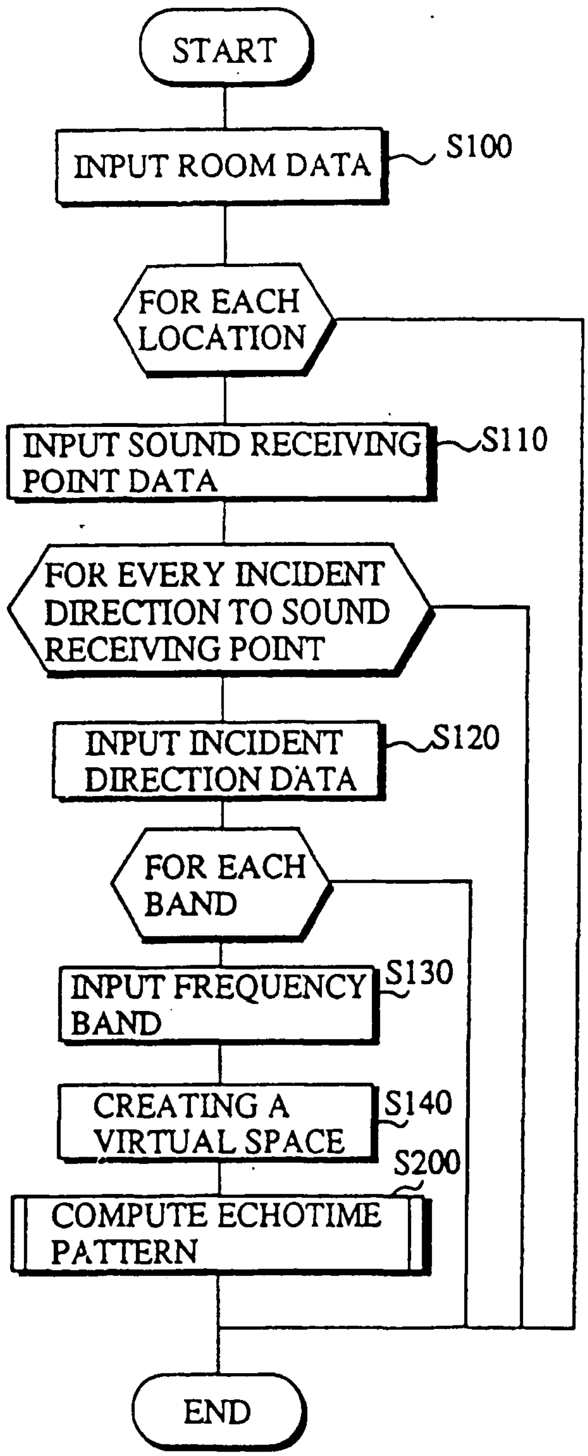

START
INPUT ROOM DATA
S100
FOR EACH LOCATION
INPUT SOUND RECEIVING POINT DATA
S110
FOR EVERY INCIDENT DIRECTION TO SOUND RECEIVING POINT
INPUT INCIDENT DIRECTION DATA
S120
FOR EACH BAND
INPUT FREQUENCY BAND
S130
CREATING A VIRTUAL SPACE
S140
S200
COMPUTE ECHOTIME PATTERN
END

Fig.4

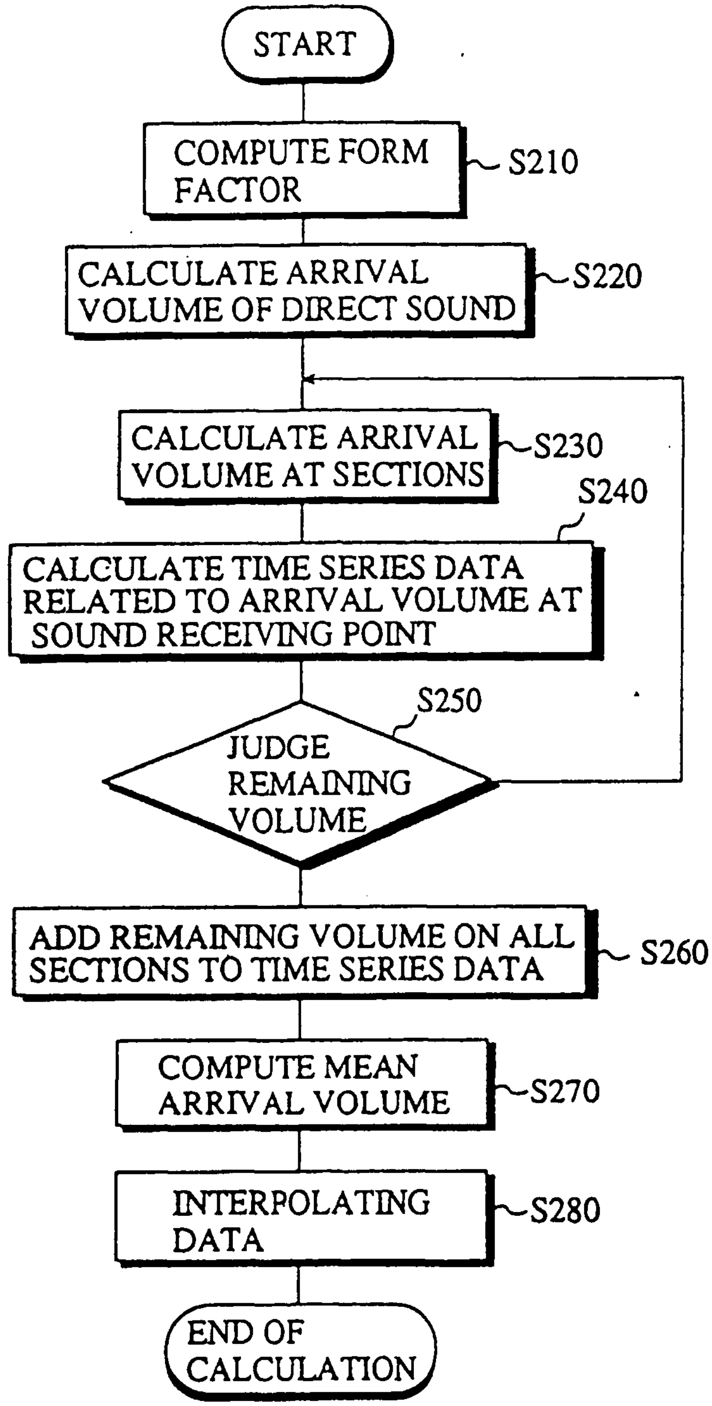

START
COMPUTE FORM FACTOR
S210
CALCULATE ARRIVAL VOLUME OF DIRECT SOUND
S220
CALCULATE ARRIVAL VOLUME AT SECTIONS
S230
S240
CALCULATE TIME SERIES DATA RELATED TO ARRIVAL VOLUME AT SOUND RECEIVING POINT
S250
JUDGE REMAINING VOLUME
ADD REMAINING VOLUME ON ALL SECTIONS TO TIME SERIES DATA
S260
COMPUTE MEAN ARRIVAL VOLUME
S270
INTERPOLATING DATA
S280
END OF CALCULATION

Fig.5

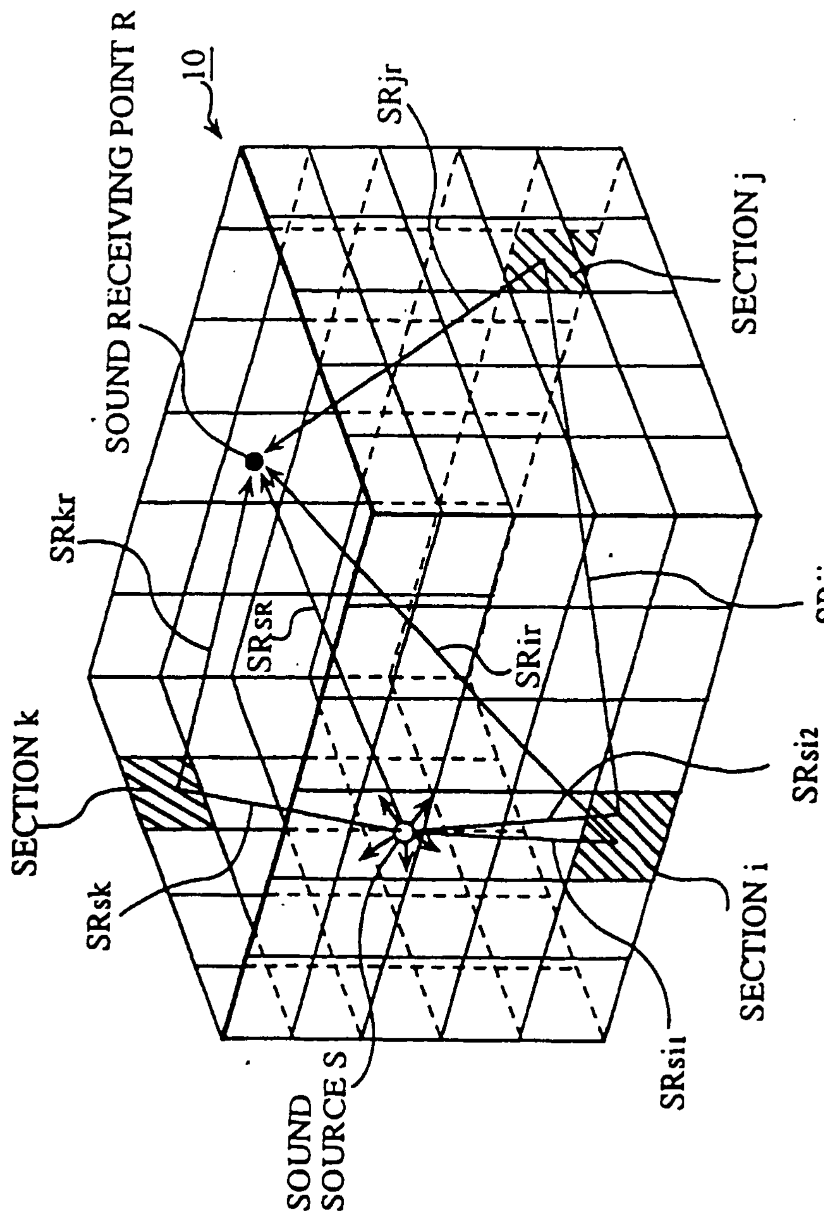

SOUND RECEIVING POINT R
10
SRjr
SECTION j
SRkr
SRsR
SRir
SRij
SECTION k
SRsk
SRsi2
SECTION i
SRsi1
SOUND SOURCE S

Fig.6

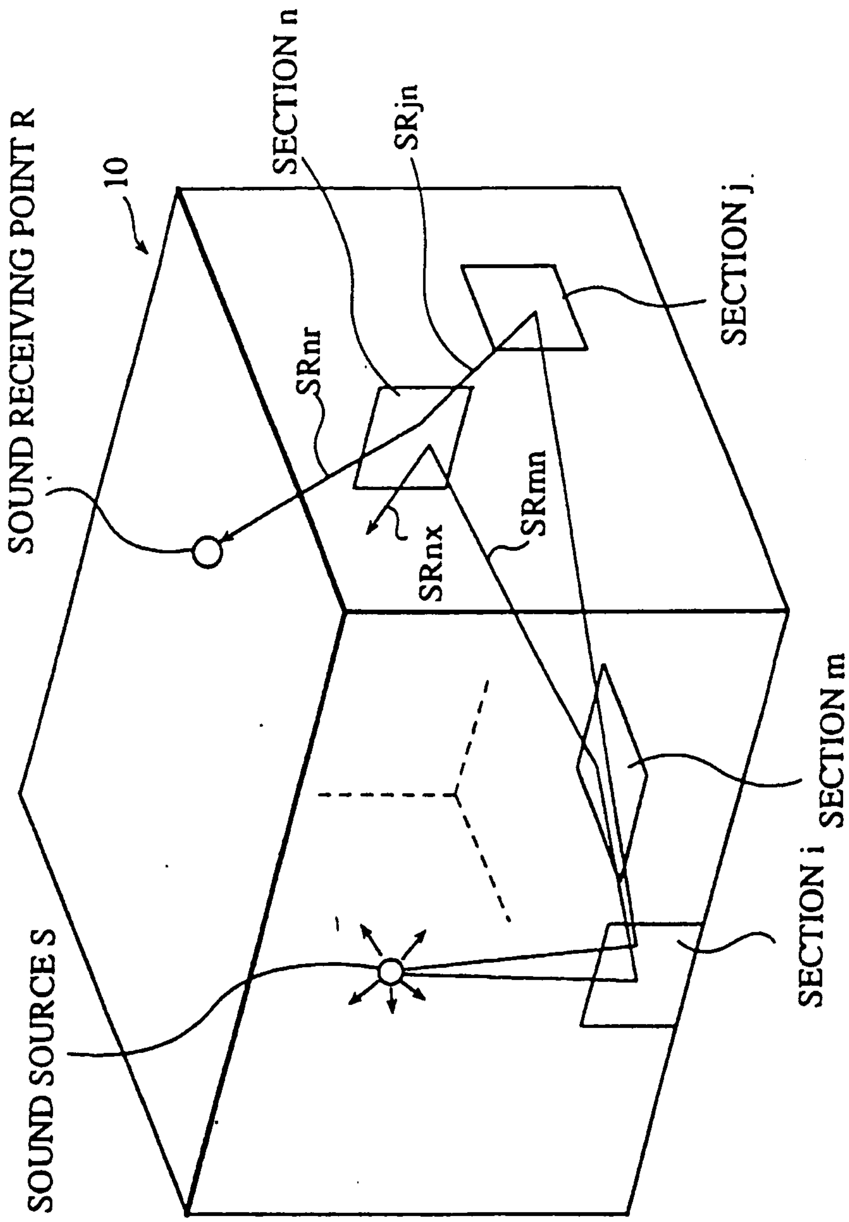

SOUND RECEIVING POINT R
10
SECTION n
SRjn
SECTION j
SRnr
SRnx
SRmn
SECTION m
SECTION i
SOUND SOURCE S

Fig.7

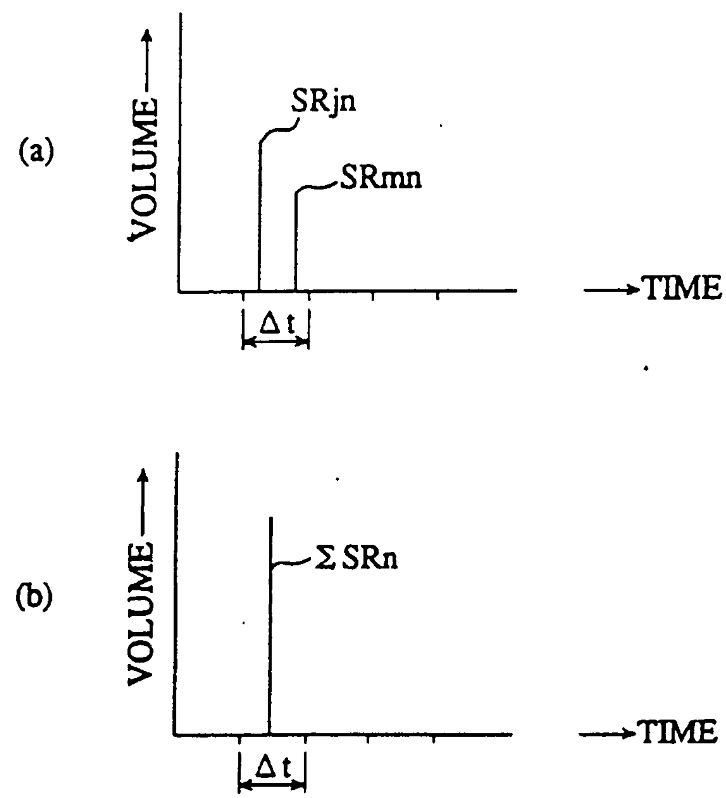


Fig.8

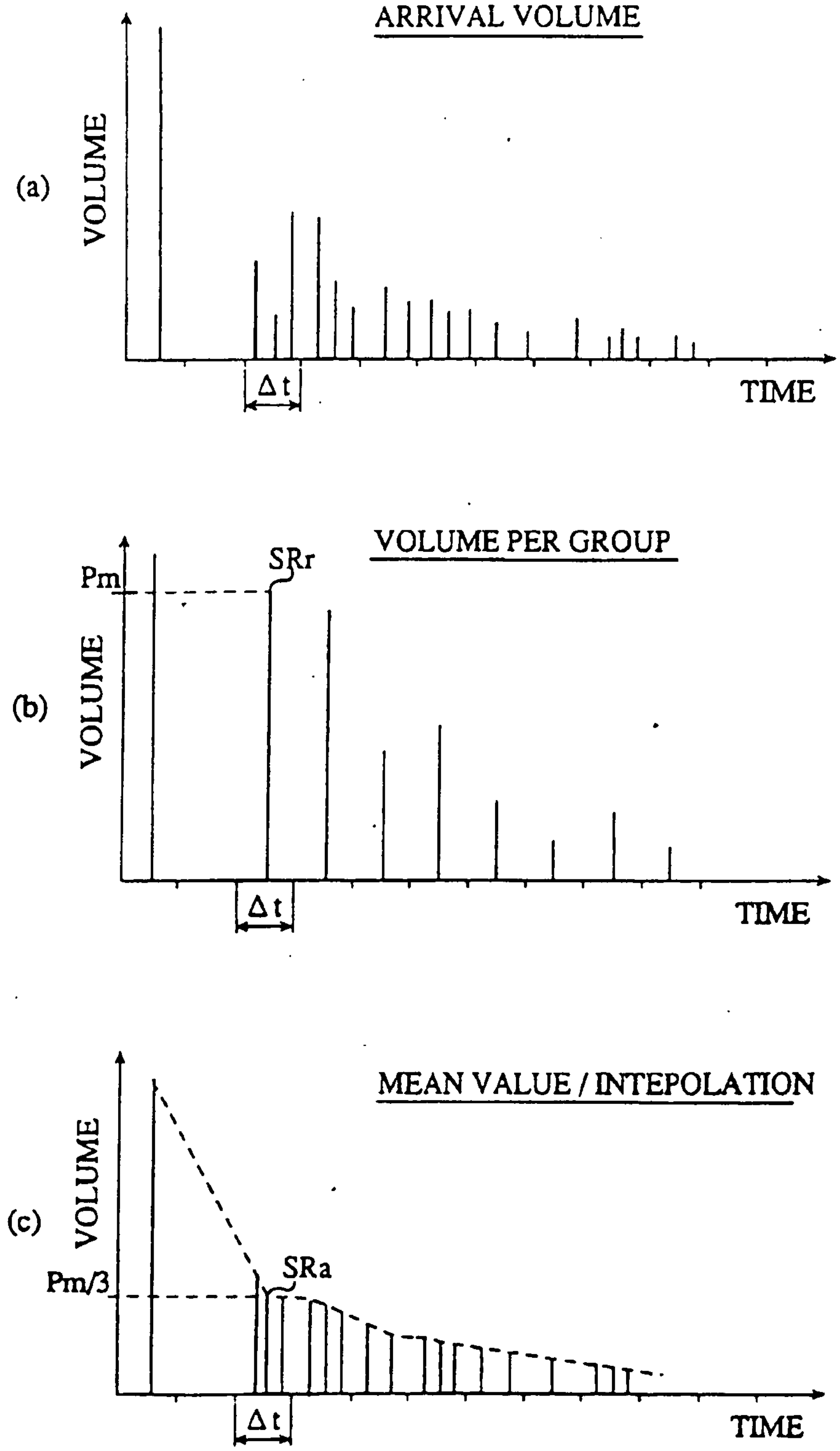

ARRIVAL VOLUME
(a)
VOLUME
Δt
TIME
VOLUME PER GROUP
(b)
Pm
SRr
VOLUME
Δt
TIME
MEAN VALUE / INTEPOLATION
(c)
VOLUME
Pm/3
SRa
Δt
TIME

Fig.9

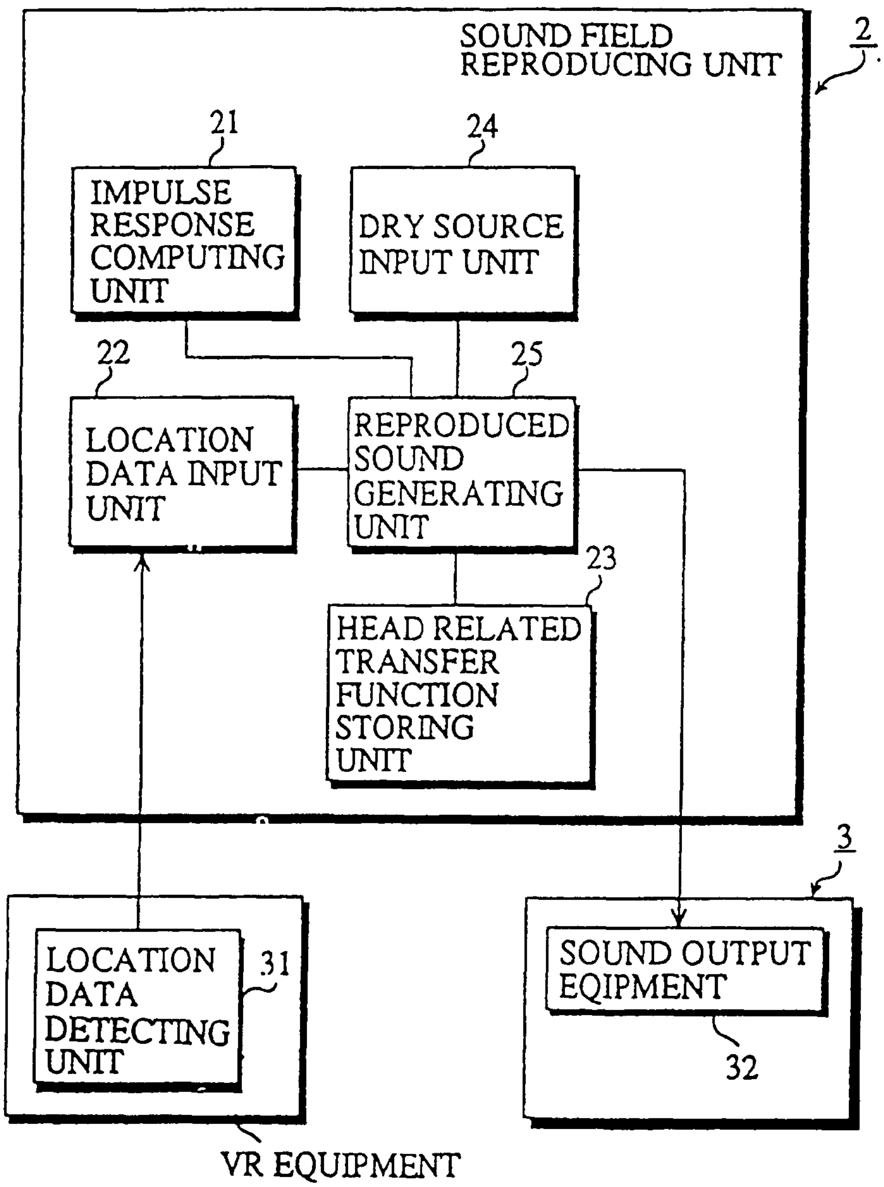

SOUND FIELD
REPRODUCING UNIT
2
21
IMPULSE RESPONSE COMPUTING UNIT
24
DRY SOURCE INPUT UNIT
22
LOCATION DATA INPUT UNIT
25
REPRODUCED SOUND GENERATING UNIT
23
HEAD RELATED TRANSFER FUNCTION STORING UNIT
3
LOCATION DATA DETECTING UNIT
31
SOUND OUTPUT EQIPMENT
32
VR EQUIPMENT

Fig.10

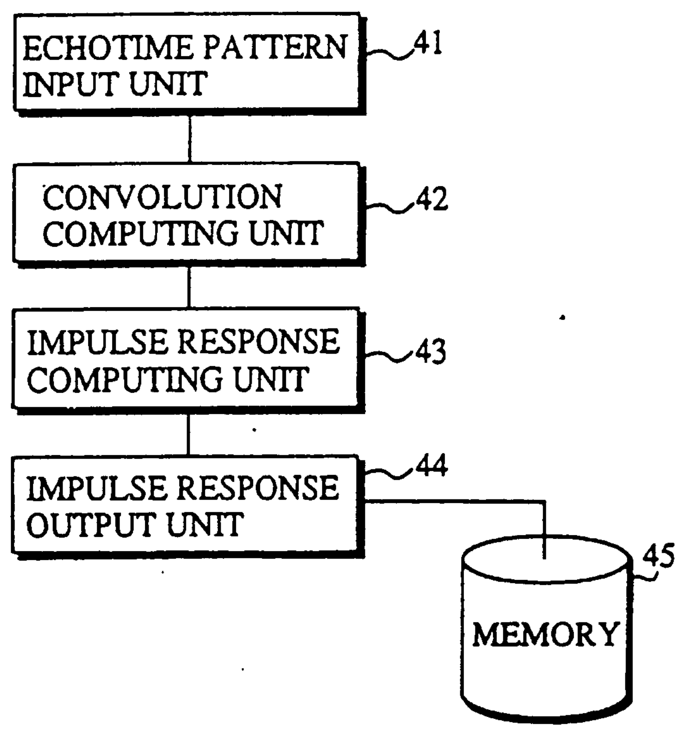

ECHOTIME PATTERN INPUT UNIT
41
CONVOLUTION COMPUTING UNIT
42
IMPULSE RESPONSE COMPUTING UNIT
43
IMPULSE RESPONSE OUTPUT UNIT
44
MEMORY
45

Fig.11

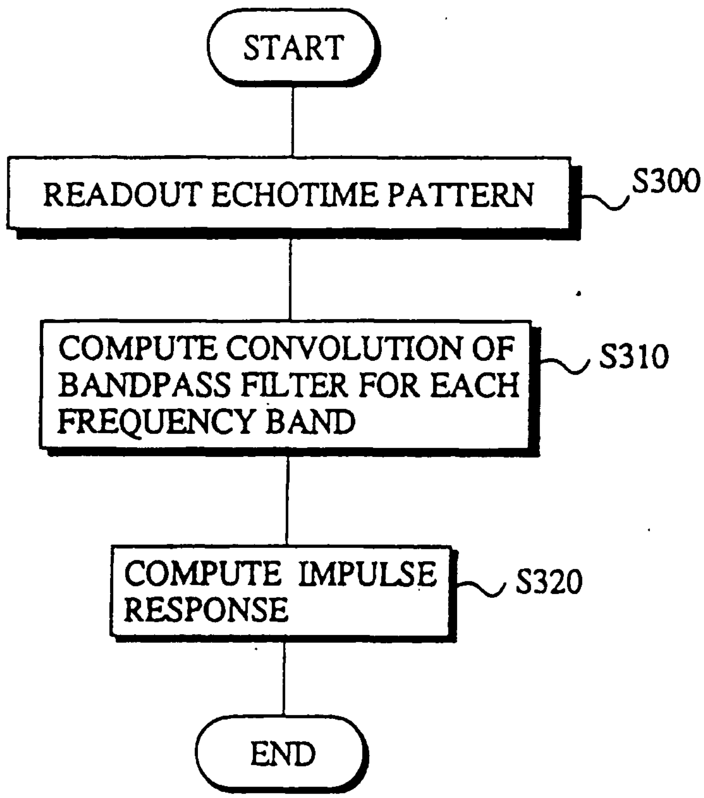

START
READOUT ECHOTIME PATTERN
S300
COMPUTE CONVOLUTION OF BANDPASS FILTER FOR EACH FREQUENCY BAND
S310
COMPUTE IMPULSE RESPONSE
S320
END

Fig.12

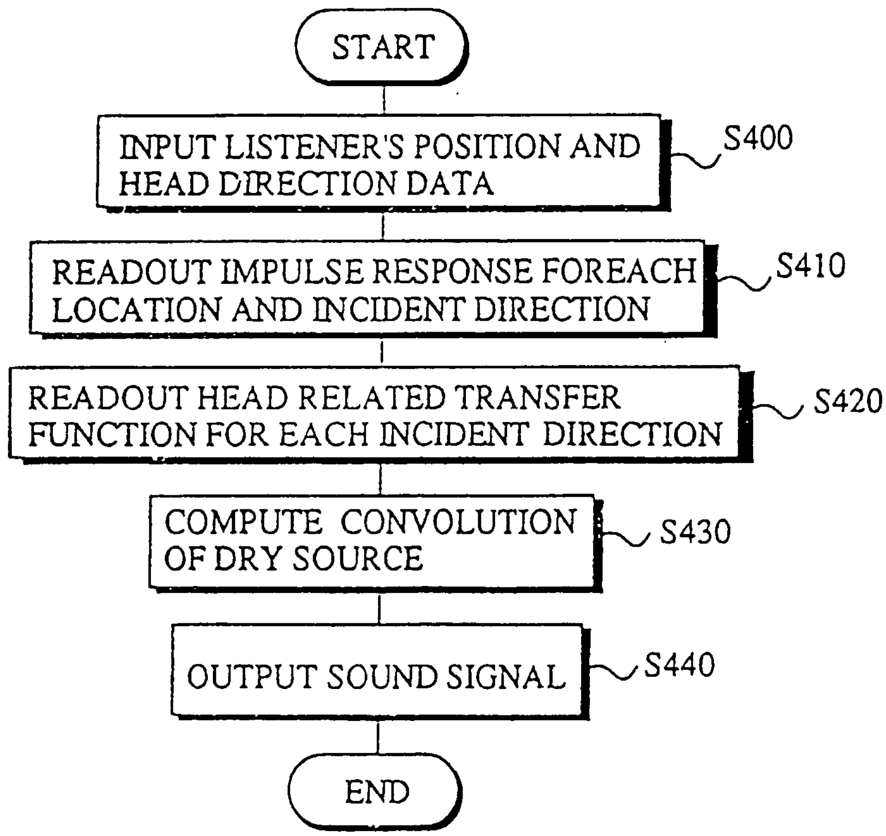

START
INPUT LISTENER'S POSITION AND HEAD DIRECTION DATA
S400
READOUT IMPULSE RESPONSE FOREACH LOCATION AND INCIDENT DIRECTION
S410
READOUT HEAD RELATED TRANSFER FUNCTION FOR EACH INCIDENT DIRECTION
S420
COMPUTE CONVOLUTION OF DRY SOURCE
S430
OUTPUT SOUND SIGNAL
S440
END

Fig.13

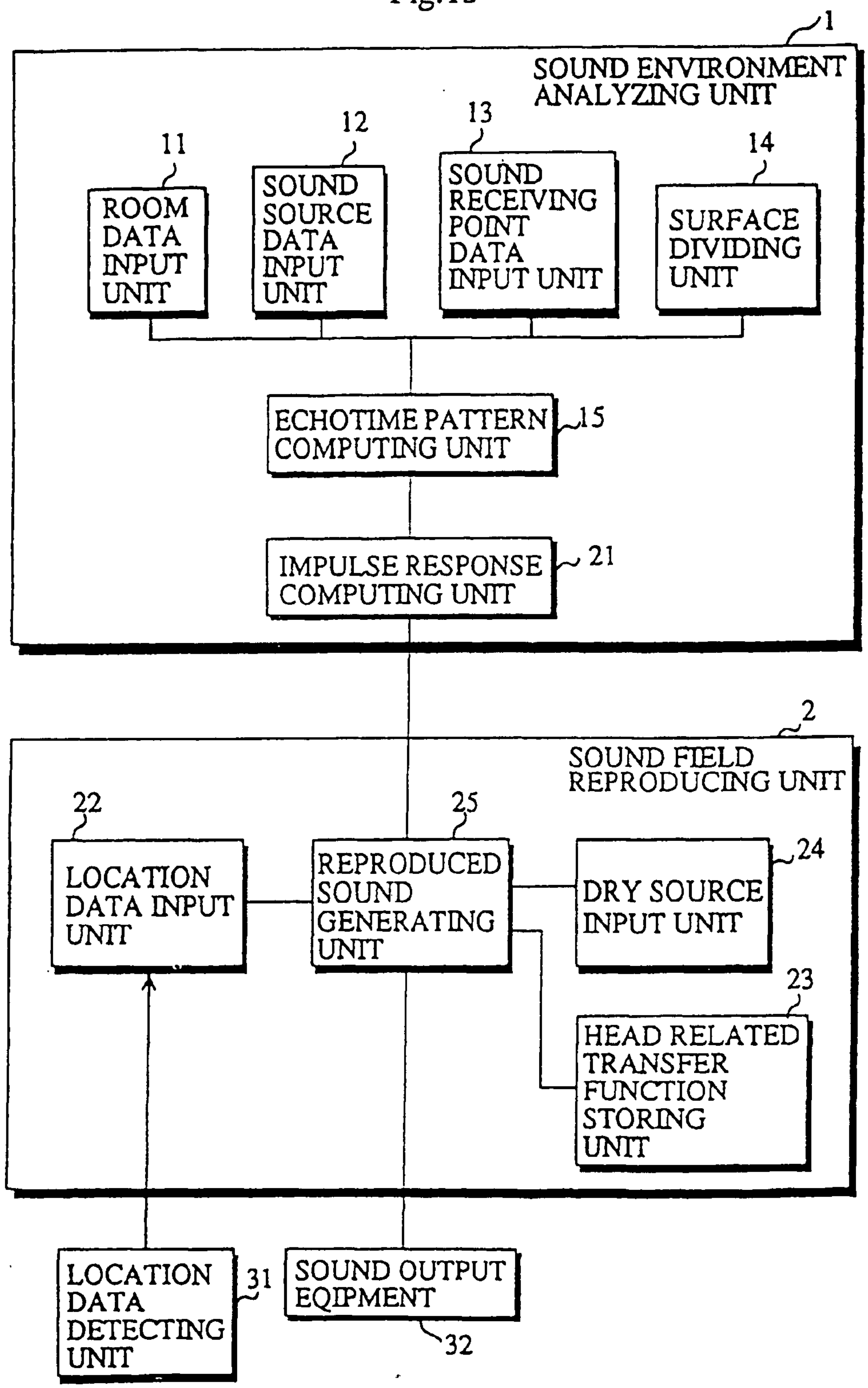
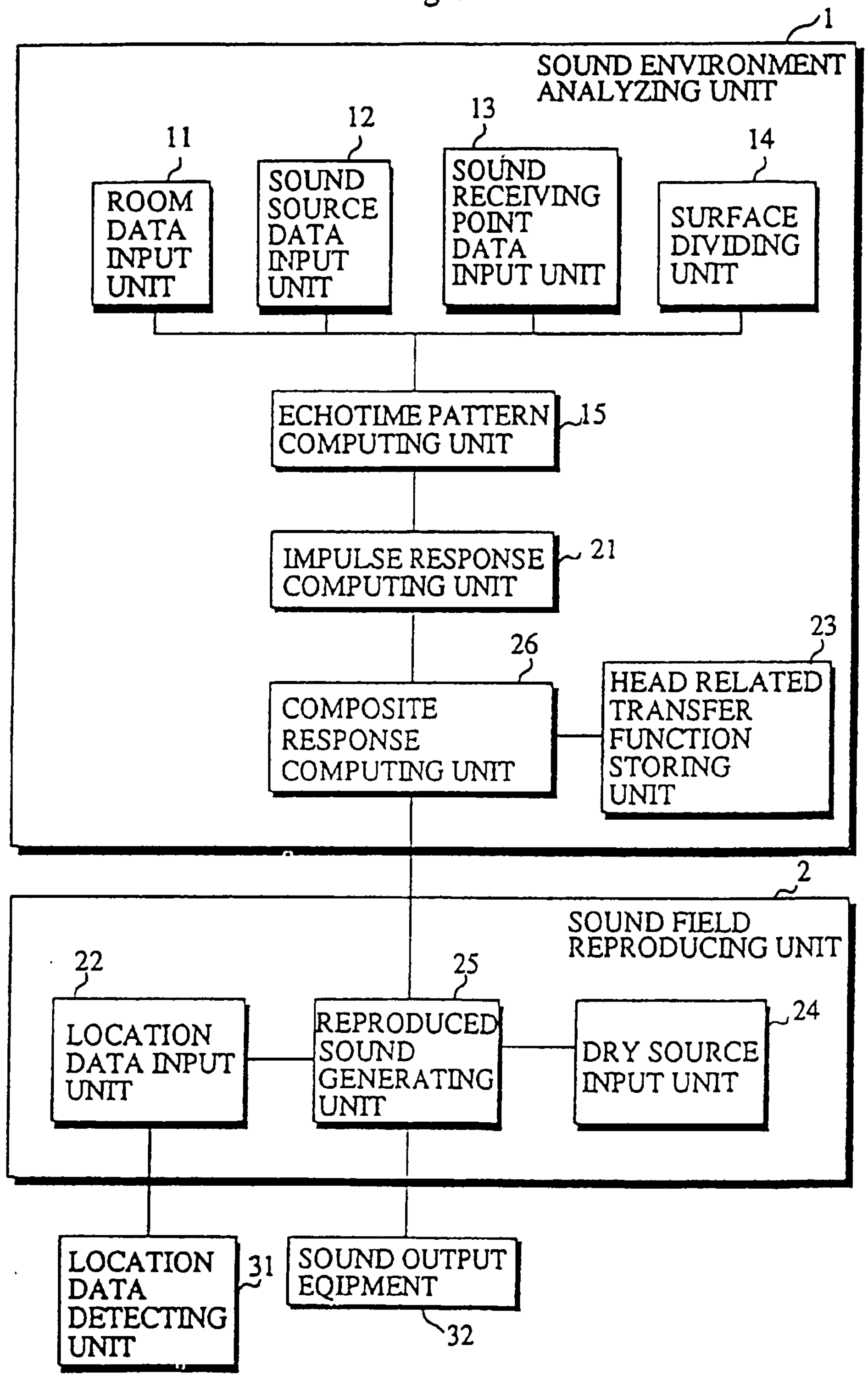

1
SOUND ENVIRONMENT ANALYZING UNIT
11
ROOM DATA INPUT UNIT
12
SOUND SOURCE DATA INPUT UNIT
13
SOUND RECEIVING POINT DATA INPUT UNIT
14
SURFACE DIVIDING UNIT
ECHOTIME PATTERN COMPUTING UNIT
15
IMPULSE RESPONSE COMPUTING UNIT
21
2
SOUND FIELD REPRODUCING UNIT
22
LOCATION DATA INPUT UNIT
25
REPRODUCED SOUND GENERATING UNIT
DRY SOURCE INPUT UNIT
24
23
HEAD RELATED TRANSFER FUNCTION STORING UNIT
LOCATION DATA DETECTING UNIT
31
SOUND OUTPUT EQIPMENT
32

Fig.14


1
SOUND ENVIRONMENT ANALYZING UNIT
11
ROOM DATA INPUT UNIT
12
SOUND SOURCE DATA INPUT UNIT
13
SOUND RECEIVING POINT DATA INPUT UNIT
14
SURFACE DIVIDING UNIT
ECHOTIME PATTERN COMPUTING UNIT
15
IMPULSE RESPONSE COMPUTING UNIT
21
26
COMPOSITE RESPONSE COMPUTING UNIT
23
HEAD RELATED TRANSFER FUNCTION STORING UNIT
2
SOUND FIELD REPRODUCING UNIT
22
LOCATION DATA INPUT UNIT
25
REPRODUCED SOUND GENERATING UNIT
24
DRY SOURCE INPUT UNIT
LOCATION DATA DETECTING UNIT
31
SOUND OUTPUT EQIPMENT
32